A METHOD OF MODIFYING LOW FREQUENCY COMPONENTS OF A DIGITAL AUDIO SIGNAL

RELATED APPLICATION

[0001] This application is a continuation of International Patent Application No. PCT/GB02/00987 filed March 6, 2002, which is here incorporated by reference in its entirety.

BACKGROUND OF THE INVENTION

Field of the Invention

[0002] The present invention relates to a method of modifying low frequency components of a digital audio signal.

Prior Art

[0003] When music is performed live using large public address loudspeakers, low-frequency vibrations (below approximately 50 Hz) are generated. These low frequency vibrations travel through the floor and are mechanically coupled to listeners in the audience, causing them to sense the vibrations through their feet and also within the chest cavity. Engineers in recording studios, when mixing music must create a musical balance such that all frequency ranges (low, mid and high) can be reproduced on a wide range of loudspeakers. When this mixed music is listened to at home, or in another setting. it is likely to be using much smaller loudspeakers than those used for live performances. These smaller loudspeakers are often inadequate at reproducing low (i.e., bass) frequencies, and this sometimes leads to a degradation in the quality of the listening experience. This problem has been addressed by using methods of enhancing the low frequency parts of an audio signal so that the quality of the music when played over small loudspeakers is improved.

[0004] A method for digitally modifying low frequency components of audio signals should satisfy the following design constraints;

a) music processed using the method should not sound unpleasant (constraint *c1*);
b) the method should not noticeably alter the tonal character of musical instruments (constraint *c2*);
c) the method should also not noticeably affect high frequency signals (constraint *c3*);

d) the method should not alter the balance of left and right channels of the stereo signal (constraint *c4*);

e) the method should not be prohibitively complex (constraint *c5*);

f) the method must not cause the maximum signal level of the digital system to be exceeded (constraint *c6*); and

g) the method must not amplify or introduce very low frequency signals that cannot be reproduced by small loudspeakers (constraint *c7*).

[0005] In digital signal processing, it is essential that the maximum signal level (i.e., unity) of the audio signal is never exceeded, otherwise extreme audible artifacts can result. For example, considering a 16-bit audio signal, the range of signal values is -32,768 to +32,767 (i.e., 2^{16} different values). If the signal value exceeds +32,767 then the signal value overflows to -32,768 rather than to its next highest value of +32,768. This creates extreme artifacts in the audio signal, and it is therefore vital to insure that the maximum signal limits are never exceeded. In analogue systems this type of signal overflow does not occur on account of the operating tolerances of devices such as valves, diodes, capacitors, transistors and resisters.

[0006] Low frequency audio signals may be enhanced using a number of techniques, some of which will now be discussed. One method of enhancing such signals is by the use of equalization wherein either one or many filters with different gain values process the signal, and lower frequency signals are given higher gain values. Thus low frequencies (and hence bass instruments) are emphasized by virtue of the fact that lower frequency energy is amplified. However, since most of the signal energy in a musical audio signal is concentrated into the lower frequencies this technique cannot amplify the low frequency signals very much before the maximum signal level in the system is exceeded, or the music sounds distorted. This method therefore breaks constraints *c6* and *c7*.

[0007] Multi-band compression is another technique that may he used to enhance bass frequencies. In this technique one or many filters are arranged in parallel, as in equalization. The output of each filter is processed by a dynamic compressor that measures the current signal level in that frequency hand and applies a gain whose amount is related to that measurement. The signal energy in each frequency band is averaged over time and the compressors are arranged to operate such that the bass

[0008] frequency bands are kept at a higher average energy than the higher

frequency bands. Bass instruments (such as bass drums, guitars etc) are therefore emphasized because the signal energy in higher on average in the lower frequency bands than in the higher frequency bands. However, since the gain of each band is independent of other bands, a musical instrument will often be noticeably tonally changed since the fundamental and harmonics of a note often falls across many frequency bands. This method therefore breaks constraint *c2*.

[0009] A device which uses a method similar to multi-band compression is disclosed in US Patent No.5,359,665 (Aphex Systems Ltd). This device is an analogue device which combines phase inverted, dynamically compressed bass frequencies with the original audio signal. It provides greater enhancement of the bass frequencies when they are at lower levels and less enhancement when they are at higher levels, thereby satisfying constraint *c6*. However, the device is not designed to work with stereo signals: if the device is used for the left and right channels of a stereo signal, the musical image will drift from side to side as the left and right gain factors are independent of one another. This breaks constraint *c4*. The device also does not employ a very low frequency cut-off filter that would prevent the breaking of constraint *c7*. For very small loudspeakers, this device would not be able to reproduce maximum level signals.

[00010] Published International Patent Application No. WO-A1-9846044 (K.S. Waves Ltd) describes an apparatus and method for bass enhancement The method uses the psycho-acoustic principle of virtual pitch, in which the pitch of a musical note is recognized by the frequency spacing and relative level of the harmonics of the fundamental of that note. The method involves taking a certain range of bass frequencies and generating artificial harmonics of these frequencies which are at a higher frequency than the fundamental bass frequencies. The original bass frequencies are not actually present in the processed signal, only the artificial harmonics, as the fundamental does not need to be present for the pitch to be recognizable. However, audio signals which are processed using this method can be musically unpleasant as, for the effect to be noticeable, frequencies in the range 100 to 300 Hz must be quite loud. This breaks constraint *c1*. Also, the apparatus has many functional components and higher-order statistical processing which breaks constraint *c5*.

SUMMARY OF THE INVENTION

[00011] An aim of the present invention is to provide a method for modifying (i.e., enhancing) low frequency components (i.e., from ~40 to ~-150 Hz) of a digital audio signal. Another aim of the present invention is to provide a method for modifying low frequency components of a digital audio signal so that the signal is suitable for playing over small loudspeakers. A further aim of the invention is to provide a method for modifying low frequency components of a digital audio signal which does not violate the constraints *c1* to *c7*.

[00012] According to the first aspect of the invention there is provided a method of modifying low frequency components of a digital audio signal as specified in the claims, and the product thereof.

[00013] According to a further aspect of the invention there is provided a computer program for implementing the method claimed.

BRIEF DESCRIPTION OF THE DRAWINGS

[00014] An embodiment of the Invention will flow be described, by way of example only, with reference to the accompanying Figures, in which:

[00015] Figure 1a shows a system diagram of the main method steps;

[00016] Figure 1b shows a system diagram of a limiter used in the method;

[00017] Figure 1c shows the output of a peak-hold circuit;

[00018] Figure 1d shows the limiter input-output characteristics; and

[00019] Figure 2 shows a combined plot of the frequency-gain characteristics of the high-pass and the band-pass filters.

DETAILED DESCRIPTION OF PREFERRED EMBODIMENTS OF THE INVENTION

[00020] Referring to Figure 1a, the first step of the method of the invention involves splitting the left (12) and right (14) channels of a digital audio signal (10) into two separate, identical signals to give a first (12a) and second (12b) left audio signal, and a first (14a) and second (14b) right audio signal. The first left (12a) and right (14a) audio signals are filtered by left (16) and right (20) high-pass filters, respectively. The left (16) and right (20) high-pass filters have the same filter characteristics.

[00021] The second left (12b) and right (14b) audio signals are passed through left (18) and right (22) band-pass filters, respectively. The left (18) and right (22)

band-pass filters have the same filter characteristics. The band-pass filters (18, 22) are designed so that they completely attenuate frequencies below a certain frequency, depending on the type of loudspeakers on which the processed audio signal is to be played. The frequency-gain characteristics of the digital band-pass filters (18, 22) for use with hi-fi speakers are shown in Figure 2. It is the relationship between the gain and the frequency of the band-pass filters (18, 22) that partly determines whether the constraints *c1* to *c7* are satisfied. It is noticeable that below the minimum frequency range of the speakers (i.e., ~50 Hz for hi-fi speakers), the band-pass filter is designed to attenuate dramatically.

[00022] The band-pass filters (18, 22) can be implemented as standard Butterworth infinite impulse response (IIR) filters, but the parameters of the filters depend upon the loudspeakers that the processed signal is to be played over. Table 1 below shows the design parameters for the band-pass filters (18,22) for different types of loudspeakers, where *fs1* is the lower cut-off frequency, *fs2* is the upper cut-off frequency, *fp1* is the lower pass-band frequency, *fp2* is the upper pass-band frequency, A*p* is the pass-band ripple and A*s* is the stop-band ripple.

Speaker Type	Lowest Frequency (Hz)	Butterworth Design Parameters *fs1*,*fp1*,*fp2*.*fs2* (Hz), A*p*,A*s* (dB)
Notebook PC	100	80, 100, 150, 180, 8, 11
Multimedia PC	80	50, 80, 120, 150, 8, 11
Hi-Fi	50	20, 50, 120, 150, 8, 11

[00023] Table 1

[00024] It is essential that the method of the invention enhances only those frequencies that can be reproduced by the loudspeakers on which the audio signal is to be played. Very low frequency signals which are below the reproduction range of small loudspeakers must therefore be removed. The amount of attenuation of these very low frequency signals is dependent on the size of the loudspeaker, thereby ensuring that constraint *c7* is not broken.

[00025] The frequency-gain characteristics of the high-pass filters (16, 20) are shown in Figure 2. These filters are also implemented as Butterworth IIR filters and,

again, the parameters of the filters depend upon the size of the loudspeakers that the signal is to be played over. Table 2 shows the design parameters for the high-pass filters (16,20) for different types of loudspeakers, where *fs* is the stop-band frequency, *fp* is the pass-band frequency, A*p* is the pass-band ripple, and A*s* is the stop-band ripple.

Speaker Type	**Butterworth Design Parameters** ***fs*, *fp* (Hz), A*p*, A*s* (dB)**
Notebook PC	160, 190 8, 14
Multimedia PC	140, 170, 8, 14
Hi-Fi	140, 170, 8, 14

[00026] Table 2

[00027] The high-pass (16, 20) and band-pass (18,22) filters divide each channel (12,14) of the stereo audio signal (10) into two separate frequency regions: a mid to high frequency region, and a low frequency region, respectively. The high-pass filters (16, 20) have an attenuation of -0.8 dB for the mid- and high-range frequencies. This has the effect of shifting the attention of the listener so that the bass frequencies/instruments become more noticeable without breaking the constraints of *c1* to *c7*.

[00028] In the next step of the method, the absolute magnitudes of the outputs of the left (18)

[00029] and right (22) band-pass filters are determined, and the signal with the largest amplitude is taken to be the control signal (28) This step is critical as it ensures that constraints *c4* and *c6* are not broken.

[00030] A level control circuit for low and medium sound is disclosed in US Patent No. 5,175,770 (Samsung). In this circuit, the left (12) and right (14) signals are added together using an adder before the main processing steps in the circuit are carried outs. However, the difficulty with this approach is that if the signal level in both the left and right input channels is larger than one half of the maximum signal

amplitude, then the signal level at the output of the adder will be greater than unity, breaking constraint c6. It can be demonstrated that in a good deal of music, common-mode signals (i.e., those signals that appear with equal amplitude in both left and right channels) are indeed extremely likely to exceed half the maximum system amplitude. This is a significant problem for fixed~point digital arithmetic.

[00031] The methods disclosed in US Patents Nos. 4,182,930 (DBX Inc.) and 4,982,435 (Sanyo) both take a sum combination of the left and right input signals thereby creating a mono bass signal that is level adjusted and added it back to the left and right input signal. The problem with this approach is that it typically affects the stereo image of the bass signals in the music, thus breaking constraint *c4*.

[00032] In the next step of the method of the invention, the control signal (28) with the largest amplitude (either positive or negative), is then passed to a limiter (34). The purpose of the limiter (34) is to limit the dynamic range of the largest control signal (28) to below the limiter threshold value T_1, and then to amplify the limited signal to increase its energy. In this way, the 0 dB level (i.e., constraint *c6*) is not exceeded, but the signal energy is amplified. The detailed operation of the limiter (34) is illustrated by the circuit shown in Figure 1b although, in practice the limiting function is carried out digitally.

[00033] The limiter (34) includes a peak-hold circuit (36), the typical output of which is shown by the dashed line in Figure 1c. This peak-hold circuit functions by efficiently calculating the L_∞ norm of the control signal (28) whose samples have been weighted according to an exponentially decaying envelope. The circuit function is described by the following operation:

[00034] $$y(z) = \left\| x(t-n) \left| K_p \right|^n \right\|_\infty$$

[00035] where t is the current time sample, n is the index of the normal (n $\in$ 0....∞), y(t) is the current peak-hold sample, x(t) is the current peak-hold input sample, and K_p is the decay coefficient. This operation smoothly tracks the peaks of the control signal (28) as they decay under the influence of the exponential envelope.

[00036] It is important that the decay time of the circuit (36) is set so that It accurately follows the amplitude of the control signal (and therefore the bass frequencies). It is also important that the circuit (36) does not clip wave peaks or hold

for an excessive amount of time after the signal level of the bass frequencies has died away. The decay dine of the peak-hold circuit (36) depends solely upon the decay coefficient, K_p and the sample rate S. The impulse response of the peak-hold circuit in decay mode is given by:

[00037] $h(n) = |K_p|^n$,

[00038] where $K_p = \exp\left(\frac{\ln(10^{-6})}{S * T_p}\right)$,

[00039] and T_p is the required delay time of the peak-hold, in seconds. The decay time is then the time taken for the peak-hold circuit to reach 1 millionth of the original amplitude. The optimum setting for T_p was found to be 0.3 (± 5%) seconds.

[00040] The output signal from the peak-hold circuit (36) is then filtered by a smoothing filter (38) which comprises a digital low-pass filter. It is important that the response of the smoothing filter (38) is set so that sharp peaks captured by the peak-hold circuit (36) are smoothed to prevent clicks in the processed output audio signal. As the smoothing filter (38) is a simple one pole low-pass filter, the feedback coefficient, K_s, is calculated as for the peak-hold circuit. Experimentation has revealed an optimum decay time, T1, for the smoothing filter (38) of 0.05 seconds. The Z-transform of the transfer function of the smoothing filter is given by:

$$H(z) = \frac{1 - Ks}{1 - Ks \cdot z^{-1}}$$

[00041] from which it can be seen that the smoothing filter (38) is normalized to have unity gain at 0 Hz. This is implemented as a standard difference equation for digital implementation.

[00042] Due to the complex relationship between the decay time of the peak-hold circuit (36)

[00043] and the transient response of the smoothing filter (38), the smoothing filter takes some time to respond to peaks from the peak-hold circuit. As a result, there are moments when the output signal energy is larger than unity, breaking constraint *c6*. To compensate for this, a small headroom attenuation H_r is subtracted from the control signal. For reasons of computational efficiency, the headroom attenuation is subtracted from the control signal in the following method step.

[00044] In the next step of the method, the amplitude of the output (100) of the smoothing filter (38) is adjusted. In analogue terms this is carried out using an

amplitude adjustment circuit which comprises a divider (42), a comparator (40), and a switch (44). The comparator (40) 'informs' the switch (44) whether to select either 1) a post-gain constant P_1, in the case where the smoothing filter output (100) is less than or equal to the threshold level T_1, or 2) a headroom compensated gain, G_c, divided by the smoothing filter output (38) if the smoothing filter output (38) is more than the threshold level.

[00045] Figure 1d shows that the effect of the amplitude adjustment circuit is to amplify the control signals that are between zero and the threshold level by a large, fixed amount, and to amplify control signals above the threshold level in a quantity inversely proportional to that level. The nature of the decreasing slope of the graph of Figure 1d is such that when the smoothing filter output (100) is equal to the headroom compensated gain G_c, the control signal energy is unchanged, and when the smoothing filter output is unity (representing 0 dB), the control signal energy is attenuated by the headroom level H_1. In other words, Figure 1d shows that when the amplitude of the bass frequency signals is very small, the bass frequencies are amplified by a large amount. When the amplitude of the bass frequencies is very large, they are attenuated by a small amount (i.e., the headroom value). There is, however, an equilibrium point at intermediate amplitude, when the bass frequencies are unchanged.

[00046] As mentioned previously for reasons of computational efficiency, the headroom attenuation implemented by combining the headroom value with the parameters G_c and P_g in the amplitude adjustment circuit. In practice, the headroom attenuation H_r is set to 3.0 dB (± 5%), although this may be adjusted depending on the type of music which is to be processed. The following three equations are used to obtain the linear values for the post-gain level P_g, the threshold level T_1, and the headroom compensated gain G_c:

[00047] $P_g = 10^{0.1|-T_{1d} - H_{rd}|}$; $T_1 = 10^{0.1 T_{1d}}$; $G_c = T_1 P_K$;

[00048] where T_{1d} is the negative threshold level in dB, and H_{rd} is the positive headroom level in dB.

[00049] The limiter output control signa1 (30) is then used to vary the amplitude of the left (46) and right (48) output signals of the band-pass filters (18,22) to give modified left (50) and right (52) band-pass filtered signals. The modified left (50) and right (52) band-pass filtered signals are then recombined with the left (54) and right

(56) high pass filtered signals, respectively, giving the bass enhanced left (58) and right (60) audio output signals.

[00050] The parameter which affects the degree to which the method of the invention satisfies constraint *c2* is the limiter threshold level T_1. In choosing the limiter threshold, care must be taken to avoid tonal alteration of the processed signal. Tonal alteration results when the relative balance of low to high-order harmonics of an instrument are affected differently, depending upon whether they fall inside or outside the bass limiter frequency range (i.e., the range of the band-pass filter). !t has been found experimentally that a threshold level T_1 of -14 dB (+ 5%) provides an optimum balance between maximum bass enhancement and minimum tonal alteration.

[00051] As the digital audio signal (10) consists of a large number of discrete samples, each step of the method is applied to every discrete sample in turn. For example, the step of calculating the absolute magnitude of the outputs of the left (18) and right (22) band-pass filters in order to find the control signal (28) is carried out on each discrete sample.

[00052] This means that the control signal (28) may be formed from the left channel in one sample, but from the right channel in the subsequent sample.

[00053] Variation may be made to the aforementioned embodiment without departing from the scope of the invention. For example, the method is also suitable for use with headphones. As headphones often have radically different frequency responses, it is possible to modify the design frequencies for the band-pass and high-pass filters such that the method can be matched to specific types of headphones such as in-ear, open or closed back headphones.

[00054] Different types of music have different dynamic characteristics. For example, dance music often has fast, staccato bass notes, and classical music has long, slow decaying notes. It is possible to provide a preset user control of parameters such as the decay coefficients K_p and K_d. The user could then select the type of music they are listening to, and have the parameters downloaded from a preset table, These parameters may be derived experimentally for different types of music or, in a further variation to the invention, higher-order statistical measurement of the dynamics of the music could allow automatic adjustment of the control parameters T_1, K_p and K_1, based on an adaptive midterm statistical measure. In

addition, the parameters of the high-pass (16,20) and/or band~pass (18,22) filters may be user selectable - parameters such as the cut-off frequency, pass-band frequency, pass-band ripple and stop-band ripple.

[00055] In a further variation, the precise mathematical relationship between the control signal greater-than-unity excursions and the peak-hold and smoothing decay times could be formulated, rather than being determined experimentally. This would allow automatic setting of the gain restoration headroom parameter, which may need to be altered if the device were to have user presets for different types of music, or if higher-order statistical processing has been use.